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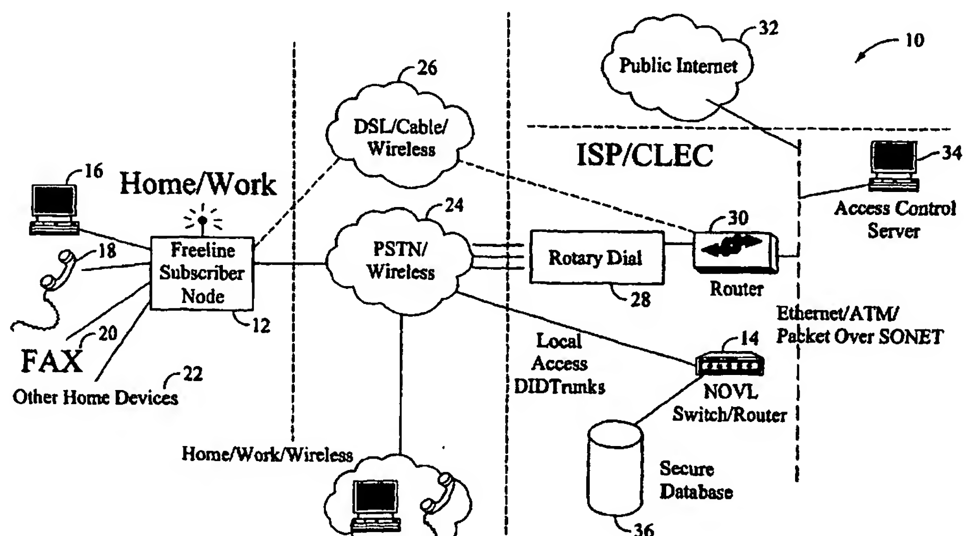
(54) Title: METHODS AND APPARATUS FOR MANAGING AN INTERLEAVED VOICE, VIDEO AND DATA COMMUNICATION SYSTEM



(57) Abstract: In one embodiment, a method for managing a system which provides one or more subscribers with a simultaneous voice and data service is disclosed. The system includes network devices and acces devices and the devices are configured to interleave voice, video, and data into subpackets within IP protocol packets for transmission and reception across at least one of public switched telephone networks (PSTN), cable data networks, Digital Subscriber Line (DSL) networks, and wireless telepone networks. The method includes remotely provisioning the network devices and access devices within the system through a computer configured for system management and tracking and monitoring system configuration, statistics and service information through the computer.

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METHODS AND APPARATUS FOR MANAGING AN INTERLEAVED VOICE, VIDEO AND DATA COMMUNICATIONS SYSTEM

CROSS REFERENCE TO RELATED APPLICATIONS

This application claims the benefit of U.S. Provisional Application No. 60/261,915, filed January 16, 2001, which is hereby incorporated by reference in its entirety.

BACKGROUND OF THE INVENTION

This invention relates generally to Internet telephone connectivity and more particularly to methods and apparatus for enabling use of the a telephone line for voice communication while online to the Internet or other on-line service.

The existing Internet connection market is based, for the most part, on dial in connectivity. This approach is cost effective and reliable for the majority of Internet customers. The major problem to be solved is the loss of the telephone line for voice communications while online to the Internet.

Several solutions exist for this problem including an extra telephone line, digital subscriber line (DSL), cable modems, Wireless Access, personal communication service (PCS) and cellular telephone accounts, and voice over internet protocol (VOIP). These options are relatively costly and or lack transparent operation from a remote caller standpoint. Also, extra physical equipment is required at the user's site. The cost for a typical email/occasional Internet user is typically not justifiable for any of these services.

An additional telephone line is at present often the best solution but typically adds an additional monthly expense to the user. Furthermore, the user must have additional wiring done to support the installation and typically must pay maintenance costs for the extra phone line.

DSL, cable modems, and wireless solutions are all relatively expensive, typically costing more per month than an additional phone line. Additional installation and equipment charges make these options too expensive in most cases for the email/occasional Internet user.

One solution that has been proposed is for the user to obtain a PCS phone and forward his or her phone to the PCS phone while on line to allow incoming voice calls to be received. This approach has many drawbacks. For example, the PCS phone may be with another family member when the calls are forwarded rather than near the computer. Furthermore, there is no easy way to get the user's telephone line forwarded and then have the forwarding removed automatically after the Internet session. Even though a forward command can be added to the Modem dial string, there is at present no easy way to undo the call forwarding. In addition, this solution is also more costly than having a second phone line for most users. Even though a user may just use the PCS phone for forwarding Internet calls, typical users still make more calls per month or use the phone for longer periods of time than provided in a typical contract.

Another proposed solution is to use VOIP. However, this approach is not transparent either to the customer or a caller to the customer. No protocol exists to handle the customer's telephone number forwarding for VOIP gateway access. The cost of the customer unit for VOIP will, by the nature of the overhead of VOIP, be relatively high. Connectivity requires that a customer install and maintain a VOIP connectivity interface, usually a headset connected to a computer using an internal multimedia audio interface. Attempts have been made to make a telephone connection to the computer using an internal adapter card and more recently a USB adapter. However, each type of connection requires a considerable amount of software and, even more importantly, relatively fast processing capability. Yet most email/occasional Internet users have computers with processing capability considered to be of a slow to medium speed.

BRIEF SUMMARY OF THE INVENTION

In one aspect, there is provided a method for managing a system which provides one or more subscribers with a simultaneous voice and data service. The system includes at least one network device and at least one access device, the devices being configured to interleave voice, video, and data into subpackets within IP protocol packets for transmission and reception across at least one of public switched telephone networks (PSTN), cable data networks, Digital Subscriber Line (DSL) networks, and wireless telephone networks. The method comprises remotely provisioning the network devices and access devices within the system through a computer configured for service management and tracking, and monitoring system configuration, statistics and service information through the computer.

In another aspect a network device which provides an interface between a local intranet and telephone voice lines for interleaved voice, video, and data within internet protocol (IP) packets is provided. The device comprises an interface to a subscriber access device, an interface to DID trunks delivering connectivity with local telephone voice lines, a connection to a local area network based intranet, a connection to a local network management computer, a memory, and a plurality of processors mapped to the memory. The processors are configured to support transmission and reception of interleaved voice, video, and data, support voice compression and decompression, and support service monitoring. The device is configured to store call detail records in the memory of calls made to the device.

BRIEF DESCRIPTION OF THE DRAWINGS

Figure 1 is a block diagram of a system configured to provide simultaneous voice, video, and data services.

Figure 2 is a diagram of an interleaved voice, video, and data internet protocol (IP) packet.

Figure 3 is a diagram of one embodiment of an access device.

Figure 4 is a diagram of another embodiment of an access device.

Figure 5 is a diagram showing one embodiment of an access device configured to interface to an ethernet network.

Figure 6 is a diagram of a network device for providing interleaved voice, video, and data services.

Figure 7 is a detailed diagram of the network device of Figure 6.

Figure 8 is a diagram of a digital signal processor (DSP) channel.

Figure 9 is a diagram of a network management system.

DETAILED DESCRIPTION OF THE INVENTION

In one embodiment, a network over voice line (NOVL) switch, sometimes referred to herein as a network device, and a subscriber node, sometimes referred to as an access device, are described. The subscriber node is configured to, in one embodiment, forward a subscriber's telephone number to an internet service provider (ISP) point-of-presence (POP). The network device is configured to handle unforwarding of the subscriber's telephone line, and supports call origination and reception to the subscriber while online to the ISP or POP, all without subscriber end processing, for truly transparent delivery of voice and Internet video and data services.

The network device (shown in Figures 6, 7, and 8 below) accomplishes the task in a manner compatible with the majority of phone systems and with minimum requirements for the subscriber, who simply attaches his modem, telephone and telephone service with a device herein called an access device (shown in Figures 3, 4, and 5 below). In one embodiment, the ISP/POP site supports the technology utilizing the NOVL switch which has minimal intrusion to the ISP's/POP's basic structure. The NOVL switch/access device system therefore provides simultaneous data, video, and telephone service that is acceptable for the majority of subscribers and is transparent to both subscribers and their callers. As used herein, simultaneous voice, video, and data, refers to internet protocol packets which include one or more of interleaved voice, video and data within subpackets as further described below. Quality of service for voice communications is not compromised as voice data is not

transferred over the Internet, but rather, voice data is received at an ISP POP where, through the NOVL switch, voice signals are transferred into local telephone lines.

In one embodiment, the access device appears transparent to the subscriber. The difference between an access device and known modems is that the access device, in one embodiment, utilizes an internal microcomputer to emulate the modem interface and interleave voice, video and data subpackets as further described below.

Figure 1 is a block diagram of a system 10 incorporating an access device 12, located at a subscribers home or work place, and a network over voice line (NOVL) switch 14, located at an internet service provider (ISP) point-of-presence (POP). Access device 12, in one embodiment, is configured to be connected to a modem of a user computer 16, a user telephone 18, a user fax 20, and other user devices 22. In addition, access device 12 is configured to be connected to a telephone service, for example, a public switched telephone network (PSTN) 24. In alternative embodiments, access device is further configured to be connected to other communications mediums 26, for example, digital subscriber line (DSL), cable modems, and wireless systems.

PSTN 24 is interfaced to a rotary dial system 28, which provides modem interfaces for the ISP. Rotary dial system 28 is connected to a router 30 which provides controlled access 34 to the Internet 32 or a Corporate Intranet. NOVL switch 14 is also connected to PSTN 24, in one embodiment, through a local access Direct Inward Dial (DID) trunk. NOVL switch 14 further interfaces to a database 36 which allows system 10 to provide secure operation, administration, maintenance, and provisioning as described below.

Access device 12 and NOVL switch 14 provide means, typically using a processor, (not shown) by which system 10 provides a user with simultaneous voice, data and video service by configuring internet protocol (IP) packet payloads with interleaved voice, data , and video for at least one of a TCP and PPP transport service. For security, the processors are configured to encrypt the IP packets, and typically use one of a 64 bit, a 128 bit, a 256 bit, and a 512 bit key encryption scheme utilizing an

industry standard encryption algorithm. Typical algorithms include Rivest-Shamir-Aldeman (RSA), Data Encryption Standard (DES), triple DES, Secure Hash Algorithm (SHA), and International Data Encryption Algorithm (IDEA) and associated message authentication procedures. Packets 50 (shown in Figure 2) and communications procedures are contemplated to meet V.34, V.42, V.42bis, V.80, V.90 and V.92 modem interface specifications.

In order to optimize the available bandwidth between subscriber access device 12 and NOVL switch 14 at the ISP POP facilities, voice and video data is compressed in a format consistent with International standards for multimedia and videotelephony services. In one embodiment, ITU-T standard recommendation H.323 is implemented given its optimization for packet networks with minimal or no guarantee of bandwidth. More specifically, for video compression, access device 12 and NOVL switch 14 will typically use at least one of the standards H.261 or H.263, for audio compression, and are configured to conform to at least one of the standards G.722, G.728, G.723, or G.729. These compression standards have been implemented in hardware, for example, application specific integrated circuits (ASICS) that process at speeds fast enough to minimize the performance impacts from the processing delays. Transmission bandwidth for voice services required by the optimized compression standards ranges from 4kbps to 8kbps which still provides available bandwidth to deliver other compressed video and data services simultaneously while the subscriber is online to the Internet. Additional specifications H.221, H.230, and H.242 are implemented in specific embodiments and address requirements for addressing, call setup and tear down, framing, multiplexing and operational functions required for interleaved voice, video, and data services.

In one embodiment, the protocols described herein for voice and data interleaving allows voice data to have a low latency, thus reducing echo problems and end-to-end delay associated with VOIP calls. Typical delays in the embodiments described herein are very small, in one embodiment less than 64 milliseconds, therefore allowing simple 512-tap echo cancellation filter technology to be utilized. Compression and decompression add most of this delay. By minimizing the latency in the packets, more processing is allowed in the compression process, increasing the

rate of compression and improving Internet speed during periods of simultaneous voice/data operation.

Figure 2 is a diagram of one embodiment of an IP packet 50 which includes interleaved digitized voice, digitized video and a data stream. Packet 50 includes a payload portion 52 and packet header portion 54. Header portion 54 typically includes a destination address for payload portion 52. Payload portion 52 is divided into multiple subportions 56 or subpackets each of which include a data field 58 and an address field 60. Packet 50 illustrates that access device 12 and NOVL switch 14 within system 10 are able to support multiple instances of voice conversations, video sessions, and data streams within the multiple subportions 56 of IP packet 50.

In the embodiment shown, packet 50 includes two digitized, independent, voice conversations 62 and 64, a digitized video data field 66, and a data field 68 transporting a packetized information stream. In a particular embodiment, processors within access device 12 and NOVL switch 14 are configurable to encrypt subportions 56 of packet 50 rather than packet 50 as a whole. Packet 50 is representative of any number of known IP packet schemes ranging in size from 64 bytes to as large as 9000 bytes. Future IP packet schemes are also contemplated as being capable of having voice, video, and data interleaved as described herein.

To prepare an IP packet for transmission, access device 12 and NOVL switch 14 are configured to enter a destination address into header portion 54 of packet 50. Further, payload portion 52 is divided into a plurality of subportions 56, or subpackets, each having a data field 58 and an address field 60. A portion of one of the voice, video, and data is loaded into each data field 58, and an identifier is loaded into corresponding address fields 60 which includes destination information and an indicator which indicates if data field 58 is loaded with digitized voice data, digitized video data, or a portion of a data stream. In one embodiment, packet 50 includes a header (not shown) which includes bits supporting priority routing within the ISP POP. In another embodiment, subportions 56 are pre-defined as to whether they will contain voice, video, or data. In alternative embodiments, the priority bits include

resource reservation protocol (RSVP), real time protocol (RTP), and other prioritizing schemes which set aside bandwidth for said IP packets routed within the ISP POP. Similar methods are used upon receipt of an IP packet, to ensuring that the digitized voice is switched to a telephone interface, and that the digitized video and data, removed from packet 50, are routed to the correct address.

While generally describing access device 12 and NOVL switch 14 as providing simultaneous voice, video, and data in a internet user and internet service provider context, such implementations are exemplary only. Other implementations exist which incorporate voice, video, and data interleaving as described herein. Examples of other networks in which interleaving may be deployed include, but are not limited to, local area networks, ethernet networks, ATM and frame relay networks. However it is important to note that all embodiments are similar in that a user is able to remain on the internet, actively transmitting and receiving data, while still being provided with telephone service.

The Subscriber Access Device

One embodiment of an access device 70 is shown in Figure 3. Subscriber device 70, according to the embodiment shown, includes an interface 72 to an external device, for example, a modem of user computer 16 (shown in Figure 1), typically through an RJ-11 or RJ45 type connection 74. Interface 72 provides logic to interface modem (not shown) to a microcontroller, or processor, 76 which provides basic control and configuration. Interface 72 further provides a DTMF encoding and decoding function and performs at least one of V.34, V.42, V.42bis, V.44, V.80, V.90, and V.92 modem interface specification thereby emulating the types found at typical ISP POP connections to allow processor 76 to communicate with the modem in user computer 16. Processor 76 is configured to perform interleaving of voice, video, and data as described above, as well as, parsing of received IP packets into the component voice, video, and data subpackets.

Access device 70 further has an interface to a communications service, for example, a telephone company dial up connection 78. Also included is a user device interface, such as a subscriber telephone connection 80. When not in use, unit

70 simply routes subscriber telephone connection 80 to telephone company connection 78. Furthermore, when the subscriber access the Internet via a personal computer or home device, modems 72 and 94 and data access processor 92 act together emulating the local dial up service offered by the telephone company, dial up modem service of the subscribers ISP, and the modem of the subscriber's PC or home device. In this mode, data packets from the subscriber are switched through the micro-controller 76 directly without experiencing any subpacket interleaving or parsing.

However when an internet connection with the ISP POP has previously been made, and the subscriber's telephone is taken "off-hook", a power level shift in circuit 90 occurs activating switch 82, which, in the embodiment shown, is a relay. Interleaving processes are activated, switching subscriber telephone connection 80 to processor 76 through a subscriber line interface circuit 84 and a compression/decompression microprocessor 86 enabling a subscriber to place and receive telephone calls as further detailed below. Subscriber line interface circuit 84 emulates a PSTN line by including a ring generator 88 and a power level shifting circuit 90. Subscriber line interface circuit 84 provides an emulation, including, but not limited to, delivering dial tone, ring generation 88, busy signal, and automatic number identification (ANI), of a telephone network. PSTN 24 (shown in Figure 1) therefore delivers the service features associated with telephone company line 78 when the access device operates in simultaneous voice and data interleave mode. Subscriber connection 80, in alternative embodiments, supports one or more of wireline telephones, wireless telephones, fax machines, and other home devices requiring Internet access. In one embodiment, processor 76 supports voice signal compression and conforms to at least one of the standards G.723 or G.729 requiring network bandwidth of less than 8kbs.

Packets of interleaved voice and data are transmitted and received through telephone company connection 78 which is interfaced to processor 76 during interleave operation. While processor 76 is configured to perform interleaving and parsing of voice, video, and data subpackets 56 within an IP packet payload, data

access processor 92 and modems 72 and 94 are configured to emulate a single user modem.

To accomplish modem emulation, access device 70 includes a data access processor 92 that meets FCC part 68 requirements and a DTMF encoding and decoding processor 94 which allows processor 76 to communicate with PSTN 24. Processor 94 further conforms to at least one of V.34, V.90, V.42, V.42bis, V.44, V.80, V.90, and V.92 modem interface specifications emulating the type found at an ISP POP connection to allow processor 76 to communicate with the modem in user computer 16. In short, access device 70 is configured to transmit and receive data and video at interface 72, transmit and receive voice communications at subscriber telephone connection 80, and transmit and receive interleaved voice and data at telephone company connection 78. Unit 70 further delivers dual modem communications linking computer modems, telephony devices, and Internet services by emulation of the modem, devices and services.

In one embodiment, subscriber line interface circuit (SLIC) 84 of access device 70 allows the user to use standard unmodified telephone devices including answering machines and other interconnecting devices, so that usage of the embodiments described herein are as simple and non-intrusive as possible. Switch 82 configures access device 70 for fail-safe operation. In addition, access device 70 is configurable to log outgoing calls, log called numbers placed from telephones that are physically connected to the access device 70, and enable the placement and reception of multiple, simultaneous, Dual Tone Multi Frequency (DTMF) telephone calls. Unit 70 is further configurable to perform a caller identification function.

Protocol for switching on and off line:

In one embodiment, microcomputer 76 is programmed to disallow the customer from accessing the Internet while the local phone is in use, as doing so would cause the current telephone connection to be disconnected. Access device 70 is configured to signal user computer 16 that the telephone line is busy so that computer 16 will notify the user of the conflict. Furthermore, to prevent hanging up a voice call by accident, access device 70 will maintain a connection after the subscriber logs off

the Internet and computer 16 disconnects from the access device. Moreover, access device 70 indicates to computer 16 that the link is terminated but holds the ISP connection until the phone call is completed.

In one embodiment, the customer can disconnect computer 16 from the Internet while online with a voice call, use other applications in computer 16, and then reconnect to the Internet in a transparent manner, in that access device 70 makes the reconnection appear to the computer software as a second Internet PPP/POP connection session making the single phone line also look as transparent to the Internet as the voice interface. Such operation is possible as unit 70 supports call origination and reception to the subscriber line while connected to at least one of the ISP POP connection and maintains call forwarding of the subscriber telephone line until the connection to the ISP POP site is broken.

In another embodiment, shown in Figure 4, an access device 100 utilizes a digital signal processor (DSP) 102 and firmware (not shown) to accomplish the functionality as described above. Access device 100, includes a serial interface 104 and a universal serial bus interface 106, providing alternative means to interconnect with the subscriber's computer. In such an embodiment, access device 100 is not transparent to a computer user, as communications have to be established with interfaces 104 and 106. A dual channel voice compression/decompression (CODEC) device 108 provides an interface to DSP 102 for subscriber telephone connection 80 and telephone company connection 78.

Figure 5 shows an embodiment of an access device 110 which operates using ethernet interfaces allowing emulation of network interfaces. A subscriber interfaces a computer to network connection 112, for example, 10 base T or 100 base T, which provides a connection to an ethernet data access processor 114 which interfaces to micro-controller 116. Device 110 communicates to an ISP, for example, by providing a second ethernet data access circuit 118 which provides a connection to an interface 120, typically one of a digital subscriber line (DSL), cable modem, or wireless GSM or CDMA network interface. Switching of ethernet packets between unit 110 and the interfaces is enabled due to interface emulation within device 110 as

described above. In such embodiments, device 110 is configured to appear as a connectionless data service using point-to-point protocol (PPP) and a TCP-IP transport layer to transmit voice, video, and data packets over an access network of an Internet service provider whose physical access services include data services delivered over CDMA or GSM wireless links, coaxial cable interconnected by transmission systems, DSL copper wire interconnected by transmission systems, or regional, ethernet data services.

Access devices 70, 100, and 110 are provisioned for a combination of services based upon the desires of the subscriber. Configuration information, performance data, call session data, and alarms are stored within a management information base (MIB) created for the access device. An internet services provider (ISP) centralized management system will routinely poll this MIB and use the data in correlating and validating the NOVL management system configuration. In one embodiment, if the subscriber adds new services such as additional telephones to a subscriber access device, for example device 70, a correlating update of the NOVL management system must be entered to ensure the telephone numbers and DID numbers will be mapped accordingly during a call placed to or from the subscriber.

The Network Over Voice Line (NOVL) Switch

Figure 6 is a diagram of a network over voice line (NOVL) system 200 which includes a network device 202 or NOVL switch (also shown as switch 14 in Figure 1) and a management system 204. Network device 202 includes a packet processing and switch matrix 206 which serves to route communications from a number of telecom interfaces 208, typically to a PSTN through a direct inward dial (DID) trunk, to a number of ISP/CLEC interfaces 210, or IP network. Control of packet processing and switch matrix 206 and interfaces 208 and 210 is provided through an operating system and network management function 212 controlling a number of processors and memory (not shown). Device 202 is an interface between an intranet and direct inward dial (DID) telephone trunks, including Signaling System #7 (SS7) interconnecting trunks, and supports simultaneous delivery of voice, video, and data transmission.

In one embodiment, device 202 is configured to support voice services by setting up a table of unused DID voice lines and associated telephone numbers, waiting for at least one subscriber access node, for example unit 70, to dial in and request one of the unused DID voice lines for call forwarding. Device 202 selects one of the DID voice lines from the table, provides a subscriber access node with a telephone number and an internet protocol (IP) address for the selected DID voice line, provides a network connection through interface 210, and sends a packet to the subscriber access node verifying a link to the DID trunk which supports voice communications. Prior to being interleaved into an IP packet payload, voice signals are compressed and digitized, in one embodiment, to conform with G.723 or G.729 ITU-T standards.

In specific embodiments interfaces 208 include, but are not limited to, at least one of T1, T3, E1, E3 ISDN PRI, SONET OC-3, SONET OC-12, SDH-1, and SDH-4, and include wireless version of these protocols. Device 202 is configured to emulate an end-office telecommunications switch, by sending ring and busy signals, sending tones for call forwarding and unforwarding, initiating call sessions upon receiving incoming and outgoing requests for voice calls, and passing through telephone service features from PSTN providers.

Furthermore, specific embodiments interfaces 210 include, but are not limited to, at least one of 10baseT, 100baseT, and Gigabit ethernet, optical OC-n and electrical STS-nc rate ATM interfaces; and optical OC-n and electrical STS-nc rate Packet Over SONET interfaces. Device 202 is configured to emulate an IP router operating as a bridge where IP packets transparently pass through device 202 without any routing decisions executed or policies being enforced.

Figure 7 is a detailed diagram of device 202. A plurality of DSP matrices 220 are interfaced to telecom interfaces 208 (previously listed) and are described in Figure 8 below. A switch matrix 222 multiplexes the digitized voice packets from DSP matrix 220 to a packet assembly and disassembly function 224 and on to interfaces 210 (previously listed). Multiple interfaces may be deployed within a single NOVL switch (device 202) and hardware redundancy is provided via a

combination of DSP switch matrices 220, the spare matrix interface 221, and the operating system 212. Upon experiencing an outage, alarms notify the operating and network management system 212, and the appropriate call sessions, including DID trunks are rerouted to spare matrix 221 for processing. In one embodiment, spare matrix 221 and interface is deployed in an Nx1 configuration where multiple interface cards and matrices are protected by a single spare.

Packet assembly and disassembly processor 224 multiplexes compressed voice, video, and data packets into subpackets in an outgoing IP packet payload and demultiplexes voice, video, and data subpackets from an incoming IP packet payload. Incoming and outgoing IP packets pass through a router matrix 226 between interface 210 and packet assembly and disassembly function 224. Router matrix 226 identifies addressable subscriber voice sessions and routes all voice, video, and data packets belonging to a common session into the packet assembly and disassembly function 224 where the voice packets are forwarded and digitized at DSP matrix 220 and mapped into data frames consistent with telecom interfaces 208, for example, PSTN and Public Wireless network services. Video and data subpackets are queued and mapped into normal IP packets, with the appropriate routing addresses, and transmitted through the local office interface 210 into the ISP's local Intranet.

Operating system and network management component 212 of device 202 provides the operational interfaces, provisioning, monitoring, and control features required for device 202 to be deployed within an ISP POP. NOVL management system 204 receives status and alarms from each of the subcomponents of device 202, which includes in the embodiment shown, interfaces 208 and 210, DSP matrices 220, PAD 224, router 226 and switch matrices 222. Upon initializing device 202, default configuration parameters , such as DID numbers, network IP addresses, and ISP dial up numbers, for all the components are transferred from NOVL management system 204 to the subcomponents listed above. A local computer bus interconnects NOVL management system 204 with the subcomponents enabling data stored in registers tracking features such as voice, video, and data subpacket interleaving and performance processing thresholds to be transferred in real time to the monitoring and alarm system for the NOVL. In case of card failures or system outages, NOVL

management system 204 captures the alarm and forwards to either a craft interface terminal or a centralized network management system of the service provider.

As the configuration is changed for the NOVL with new interface cards added or replaced, a provisioning component of NOVL management system 204 enables either a local operator or remote operator to modify and upload the new system configuration. All appropriate alarms and monitoring points are assigned upon completing the configuration updates.

Packets that pass through router matrix 226 contain origination and destination addresses for Internet World Wide Web sites that are captured and stored within a relational database created for the subscriber. Each of the Web addresses provide a historical reference simultaneously describing the call detail of the voice call and the details of the Internet session.

Figure 8 is a diagram of an individual DSP channel 250 from DSP matrix 220. DSP channels 250 include a dual-tone multiple frequency (DTMF) transmit circuit 252, a DTMF receive circuit 254, a voice compression codec 256, telephony connection logic 258 and a memory array 260 for queueing data, completing call setup procedures, and processing voice calls. An interface circuit 262 creates data frames compliant with industry standard telephony interfaces which are described above.

DSP channel 250 is provisioned with interface cards consistent with the local access DID trunks provisioned into an ISP POP. In addition, DID numbers are provisioned into network management platform 212 (shown in Figure 6), and are dynamically assigned on a per call basis. The DID numbers are stored within memory array 260. Compression and decompression of voice signals, as well as, echo cancellation is completed by processor 256. Telephony logic processor 258 completes in-band and out-of-band signaling for setting up and tearing down circuits associated with voice calls. Out-of-band signaling is compliant with the industry standard SS7. In one typical embodiment of the DSP channel, T1 DID trunks are installed at the ISP POP and interconnected with T1 interface cards deployed within NOVL switch 202. In one embodiment, block of three hundred DID numbers are assigned to DSP matrix

220 and these are stored within memory array 260. In this same embodiment, voice CODEC 256 is configured to comply with H.323 specifications utilizing G.723 voice compression thereby delivering a bit stream that is packetized with an address and forwarded to switch matrix 222 (shown in Figure 7) for processing.

An application program ties the components of network device 202 together and controls their operation. In one embodiment, the NOVL application starts by setting up a table of DID (direct inward dial) numbers offered by a PSTN service provider and mapping the DID numbers to incoming telephone trunks deployed by the PSTN provider. In one embodiment, setting up the table includes installing a number of modem lines that are proportionally mapped to a number of voice lines, based upon an anticipated call volume. Network device 202 then waits for access device 70 to call and request one of the DID numbers and trunks to be assigned and activated with call forwarding. When such a call arrives, one of the DID numbers and associated trunks are selected from routing tables using a network over voice line (NOVL) provisioning system. Unit 70 is provided with an Internet access telephone number and an internet protocol address for the assigned node of network device 202. A network connection is established by completing a modem connection between unit 70 and a local Internet access port, and a message is sent to unit 70 verifying voice services are activated for simultaneous voice, video, and data. When a telephone call is received at a DID number, network device 202 sends a packet to access device 70 switching to simultaneous voice, video, and data.

When packets 50 of interleaved voice, video, and data are received at unit 70 or network device 202, routing addresses for voice subpackets 56 are identified, the voice information is decompressed into voice signals and digitized. Similarly, routing addresses of video subpackets 64 and data subpackets 66 are identified and new IP packets are constructed for transmission into a local area network which provides access to the Internet.

When voice call sessions have ended, network device 202 terminates the call at the DID number, and transmits one or more packets to the subscriber access node, unit 70, to switch access device 70 to a data only service.

Management Systems for Voice, Video, and Data Service Delivery

Industry standard practices apply to the operation, administration, maintenance and provisioning of a simultaneous voice, video, and data service delivery system. These models are applied to the unique architecture elements that enable Internet Protocol to serve as a virtual access channel carrying compressed, voice packets interleaved with video and data. Several different data services may act as the link between subscriber and PSTN voice network including PSTN dial-up, Internet servcies, cable modem data services, DSL services, and wireless access services.

As shown in Figure 9, a topology of network management system 300 includes subscriber access nodes 310 (referred to as access devices 70, 100, and 110 in Figures 3, 4, and 5 respectively), located at the home or work place of the subscriber, NOVL network nodes 320, (referred to as devices 202 in Figures 6 and 7), located within an ISP POP, a centralized operations system 340 located at the operations center of the ISP 350 with a secure relational database 330 and protocol interfaces that enable both real time and non-real time information to be passed between system.

The entire network of devices are defined within a MIB hierarchical tree structure that includes service ports on the access devices 310 through the interface cards on NOVL network devices 320 and operations system interfaces 340. In one embodiment, all elements of the system are remotely accessible through a secure, virtual private network, TCP/IP connections or direct dial-up, access ports. Password access to elements and packet encryption is deployed to increase the security of the virtual private network linking the devices. The sub-element MIBs for each of the system components are integrated within the centralized management systems 340 and 350. Industry standard management system protocols including at least one of Simple Network Management Protocol (SNMP), SNMP v2, CMIP, UDP, TCP/IP, ethernet, and asynchronous transfer mode (ATM), are used to remotely access MIB information and remotely provision changes and updates to the configuration of the system elements.

In several example embodiments, MIB definitions are provided for all system elements including access device interfaces V.34, V.42, V.42bis, V.80, V.90, and V.92 modem interfaces; network device interfaces 10baseT, 100baseT, and Gigabit ethernet; optical OC-n and electrical STS-nc rate ATM interfaces; and optical OC-n and electrical STS-nc rate Packet Over SONET interfaces; and T1, T3, E1, E3 ISDN PRI, SONET OC-3, SONET OC-12, SDH-1, and SDH-4, and include wireless version of these framing protocols; and core elements such as the switching and routing matrices, DSP processors, and packet multiplexors and processors.

The MIB variables are stored for the various elements within memory based registers that are polled via a centralized management system using network protocols described previously.

The network operations systems 340 and ISP centralized management systems 350 typically are created using industry standard operating systems UNIX, LINUX and Windows.

In one embodiment, a new subscriber deploys access device at the home or workplace and a new customer record is created within a relational database located in the operations system 340 and ISP network management system 350. Service is provisioned by creating new records identifying the telephone number of the suscriber, linked with the ISP, and local dial up number for the participating ISP. Port configurations for the access device deployed by the subscriber are also loaded within the new records created for said subscriber.

Within the ISP, NOVL network devices 320 are deployed with DSP matrices built into specific interface cards that enable the PSTN to deploy DID trunks between the ISP and the PSTN end office circuit switch. Ports for the NOVL network device 320 are deployed proportionally to the number of ports available for dial-up use by the ISP. The appropriate voice compression and decompression algorithms such as G.723 or G.729 are activated in pairs between the access devices and appropriate network devices.

As new calls are either placed or received, NOVL network devices 320 will create a new call detail including time of day, originating and terminating telephone numbers, and user account. In addition, given each data packet is demultiplexed and parsed within the network device, the originating and destination IP addresses are extracted and mapped into the header for an IP packet destined for the Internet. As a result, IP addresses of World wide Web Sites accessed by the subscriber can be captured and stored within the same call detail record that has been created while a user is on the phone. Furthermore, voice packets can be copied and sent to a test or security port to meet the federal guidelines for service oversite and access. The duplicate voice packets are forwarded to a unique interface port 208 (shjown in Figure 7) on NOVL network devices 320 where the voice signals are reconstructed meeting federal regulatory and homeland security requirements. All of the data captured is done on a real time basis to ensure minimal delays result from the call processing and storage functions.

In a similar approach, performance thresholds and alarms defined within the MIB of each element are captured into a centralized data management system. This information is collected and stored for future use. This configuration, monitoring alarms, and performance data is collected and summarized for the ISP administration and billing personnel to use as needed.

As used herein, the terms processor, microcontroller and micro-computer broadly mean microprocessors, computers, microcontrollers, reduced instruction set circuits (RISC), application specific integrated circuits (ASICs), programmable logic controllers (PLCs), and all other programmable circuits capable of executing the methods described herein.

The embodiments described herein provide true simultaneous connection to voice and internet, as the voice and data are interleaved within the communications packets. Other known systems cannot provide true simultaneous communications as those systems are configured to put the internet on hold to take care of voice, and allow a reconnection to the internet, that is, the point-to-point protocol (PPP), which connects a computer to internet, is put on hold. While the

invention has been described in terms of various specific embodiments, those skilled in the art will recognize that the invention can be practiced with modification within the spirit and scope of the claims.

WHAT IS CLAIMED IS:

1. A method for managing a system which provides one or more subscribers with a simultaneous voice and data service, the system including at least one network device and at least one access device, the devices configured to interleave voice, video, and data into subpackets within IP protocol packets for transmission and reception across at least one of public switched telephone networks (PSTN), cable data networks, Digital Subscriber Line (DSL) networks, and wireless telephone networks, said method comprising:

remotely provisioning the network devices and access devices within the system through a computer configured for service management and tracking; and

monitoring system configuration, statistics and service information through the computer.

2. A method according to Claim 1 wherein remotely provisioning the network devices and access devices comprises using network management protocols to download configuration changes to the network devices and access devices within the system.

3. A method according to Claim 1 wherein monitoring system configuration, statistics and service information comprises using a secure polling process to collect performance statistics stored in registers from the network devices and access devices within the system.

4. A method according to Claim 1 wherein the system includes a database, wherein system configuration, statistics and service information comprises:

logically mapping interface port addresses of subscriber access devices with at least one telephone number; and

storing the mapped addresses and telephone numbers in a relational structure within the database.

5. A method according to Claim 4 further comprising configuring the database for secure access.

6. A method according to Claim 4 further comprising storing telephone call records per subscriber for voice calls placed through the network devices and access devices and IP addresses of Internet World Wide Web sites accessed by the subscriber during a voice call.

7. A method according to Claim 6 wherein storing telephone call records comprises storing telephone call records on a real time basis.

8. A method according to Claim 1 wherein remotely accessing register settings within the network devices and access devices comprises downloading register settings to monitor and maintain configuration control of network devices and access devices within the system.

9. A method according to Claim 1 further comprising accessing system usage information and voice packets through a non-obtrusive, secure and transparent interface access.

10. A method according to Claim 9 wherein the secure and transparent interface access is configured to enable real time, simultaneous access to at least one of current call sessions and download records of previously completed calls per subscriber.

11. A network device which provides an interface between a local intranet and telephone voice lines for interleaved voice, video, and data within internet protocol (IP) packets, said device comprising:

an interface to a subscriber access device;

an interface to DID trunks delivering connectivity with local telephone voice lines;

a connection to a local area network based intranet;

a connection to a local network management computer,

a memory; and

a plurality of processors mapped to said memory, said processors configured to support transmission and reception of interleaved voice, video, and data, voice compression and decompression, and service monitoring, said device configured to store call detail records in said memory of calls made to said device.

12. A network device according to Claim 11 wherein said device is configured to store internet session detail records in said memory of IP addresses of Internet World Wide Web sites accessed through said device.

13. A network device according to Claim 11 wherein said device is configured to store call detail records in said memory of calls received by said device, the detail records including a number from which said device was dialed.

14. A network device according to Claim 11 wherein said connection to a local network management computer provides access to said memory of said device.

15. A network device according to Claim 14 wherein a portion of said memory is configured as a relational database.

16. A network device according to Claim 11 wherein said memory is configured with at least one of a subscriber access number, DID numbers, network IP addresses, and ISP dial up numbers for at least one a subscriber, a PSTN, and an ISP.

17. A network device according to Claim 11 wherein said device is configured with a management information base and associated protocols thereby allowing configuration updates for said device and statistics retrieval from said device through said connection to a local intranet.

18. A network device according to Claim 17 wherein the protocols are at least one of Simple Network Management Protocol (SNMP), SNMP version 2 (SNMPv2), CMIP, UDP, TCP/IP, ethernet and asynchronous transfer mode (ATM).

20. A network device according to Claim 11 wherein for service monitoring said device is configured to:

poll an ISP intranet through said connection to a local intranet to determine service outages at the ISP;

poll and retrieve management information database information for access devices;

poll and retrieve management information database information for components within the network devices including interface cards, switch and routing matrices, packet multiplexing processors, DSP processors, and operating system elements; and

store all retrieved management information within said memory.

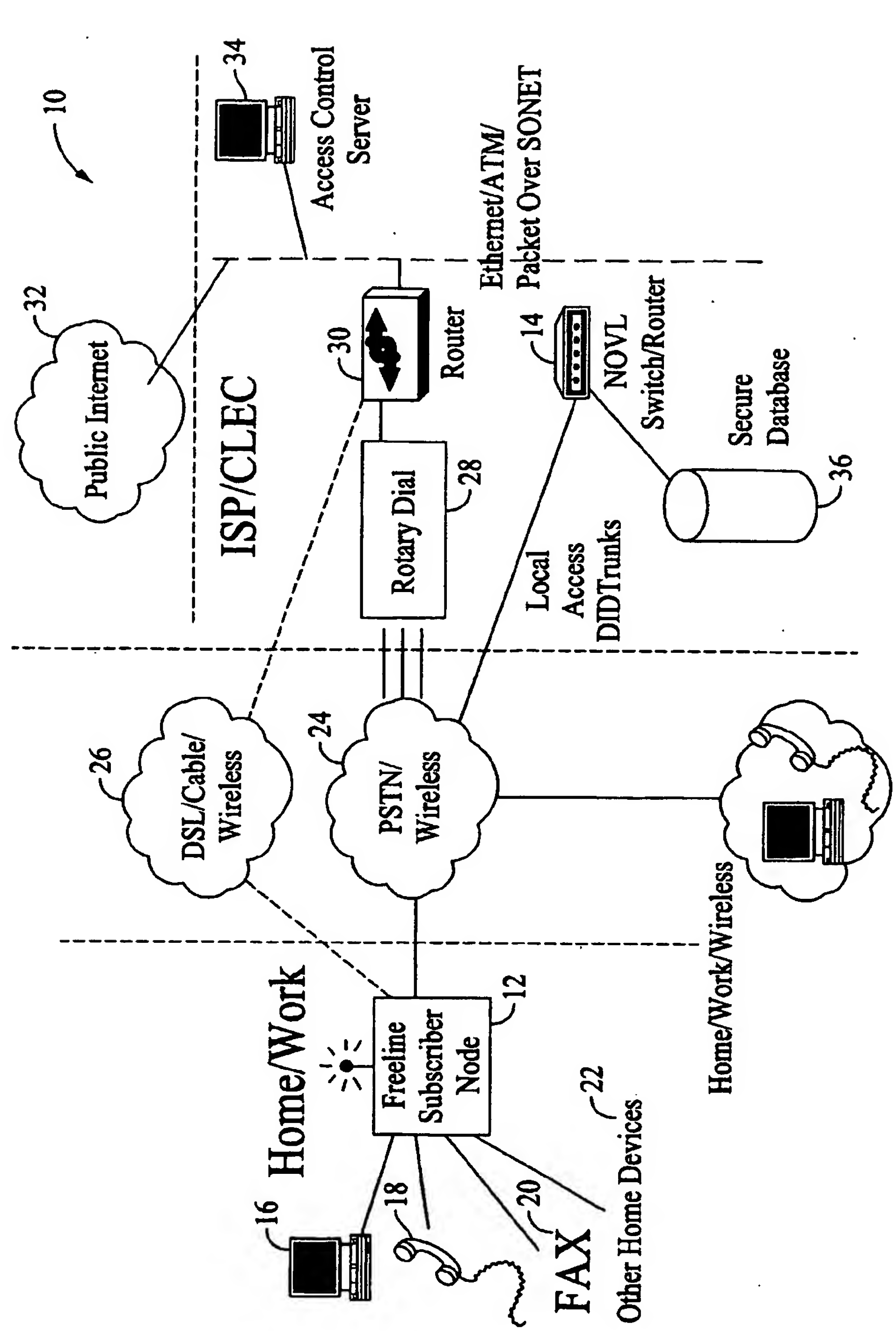

FIG. 1

SUBSTITUTE SHEET (RULE 26)

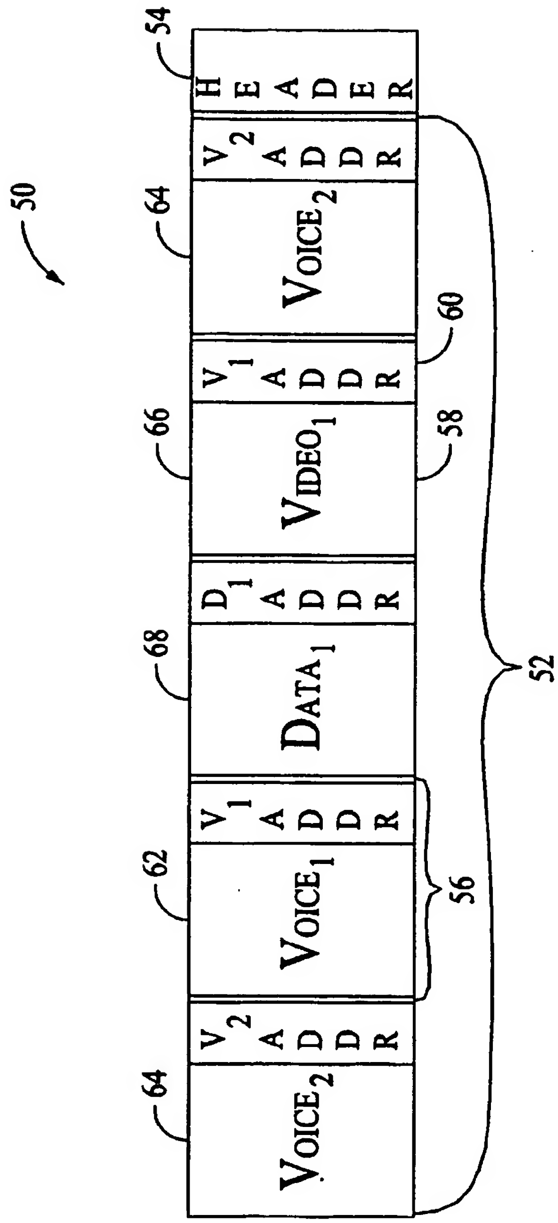
50
54
HEADER
V2 ADDR
VOICE2
64
V1 ADDR
60
VIDEO1
66
58
D1 ADDR
DATA1
68
52
V1 ADDR
VOICE1
62
56
V2 ADDR
VOICE2
64

FIG. 2

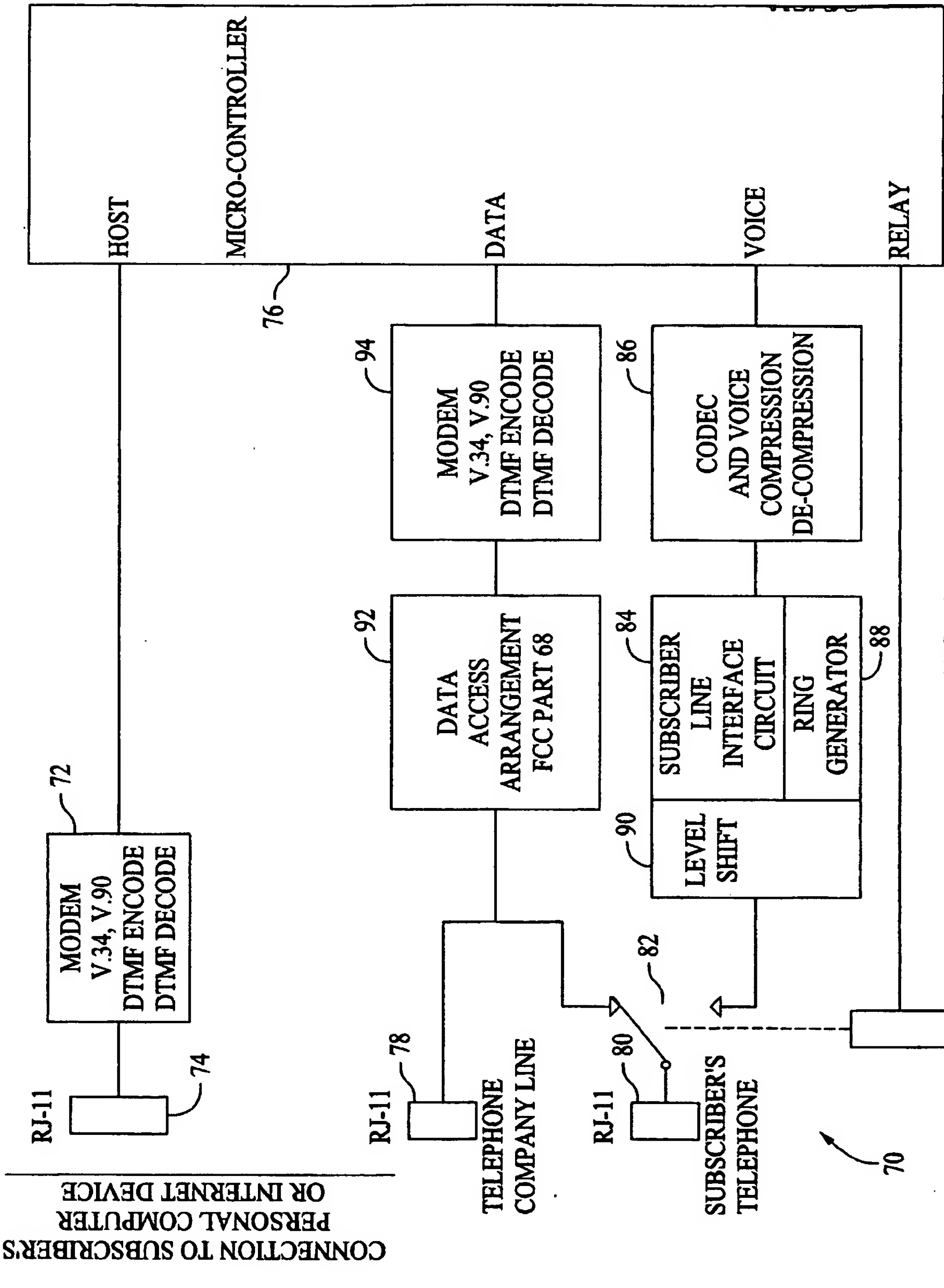
MICRO-CONTROLLER
HOST
DATA
VOICE
RELAY
76
MODEM
V.34, V.90
DTMF ENCODE
DTMF DECODE
94
CODEC
AND VOICE
COMPRESSION
DE-COMPRESSION
86
DATA
ACCESS
ARRANGEMENT
FCC PART 68
92
SUBSCRIBER
LINE
INTERFACE
CIRCUIT
84
RING
GENERATOR
88
LEVEL
SHIFT
90
MODEM
V.34, V.90
DTMF ENCODE
DTMF DECODE
72
RJ-11
74
RJ-11
78
TELEPHONE
COMPANY LINE
RJ-11
80
SUBSCRIBER'S
TELEPHONE
82
70
CONNECTION TO SUBSCRIBER'S
PERSONAL COMPUTER
OR INTERNET DEVICE

FIG. 3

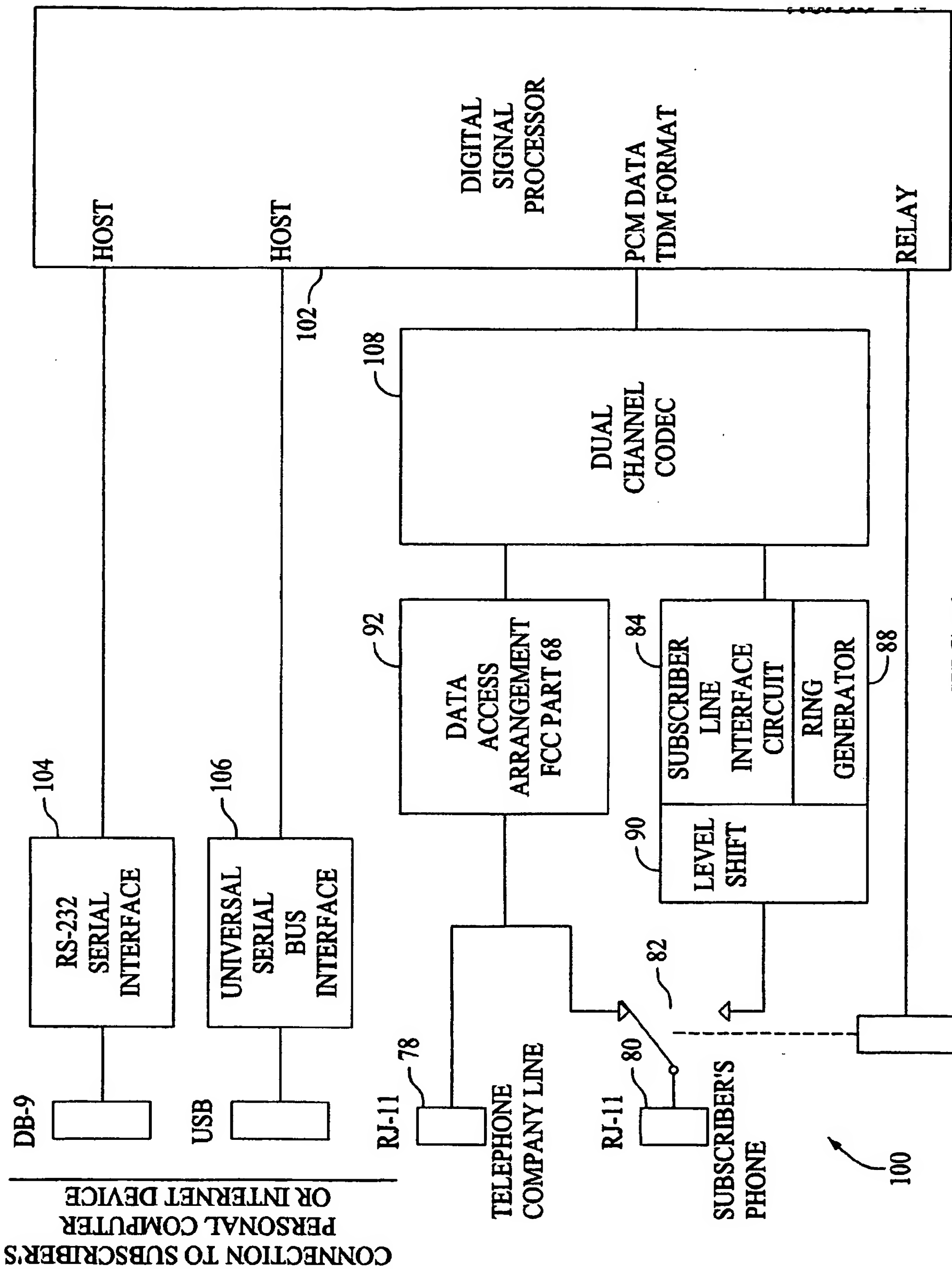
DIGITAL SIGNAL PROCESSOR
HOST
HOST
PCM DATA TDM FORMAT
RELAY
102
108
DUAL CHANNEL CODEC
92
DATA ACCESS ARRANGEMENT FCC PART 68
84
SUBSCRIBER LINE INTERFACE CIRCUIT
RING GENERATOR
88
90
LEVEL SHIFT
104
RS-232 SERIAL INTERFACE
106
UNIVERSAL SERIAL BUS INTERFACE
DB-9
USB
CONNECTION TO SUBSCRIBER'S PERSONAL COMPUTER OR INTERNET DEVICE
RJ-11
78
TELEPHONE COMPANY LINE
RJ-11
80
SUBSCRIBER'S PHONE
82
100

FIG. 4

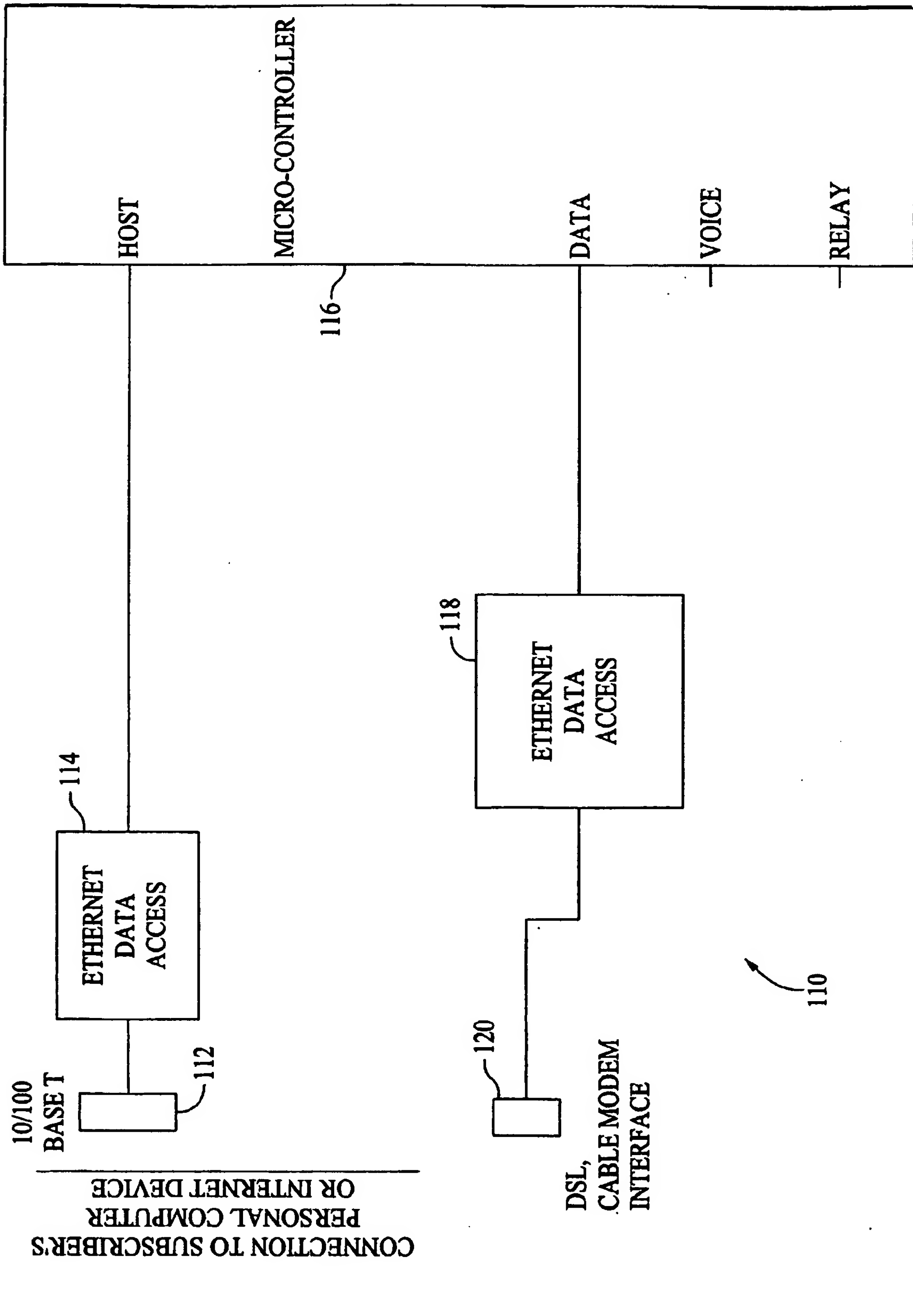
MICRO-CONTROLLER
HOST
DATA
VOICE
RELAY
116
114
ETHERNET DATA ACCESS
118
ETHERNET DATA ACCESS
10/100 BASE T
112
120
DSL, CABLE MODEM INTERFACE
110
CONNECTION TO SUBSCRIBER'S PERSONAL COMPUTER OR INTERNET DEVICE

FIG. 5

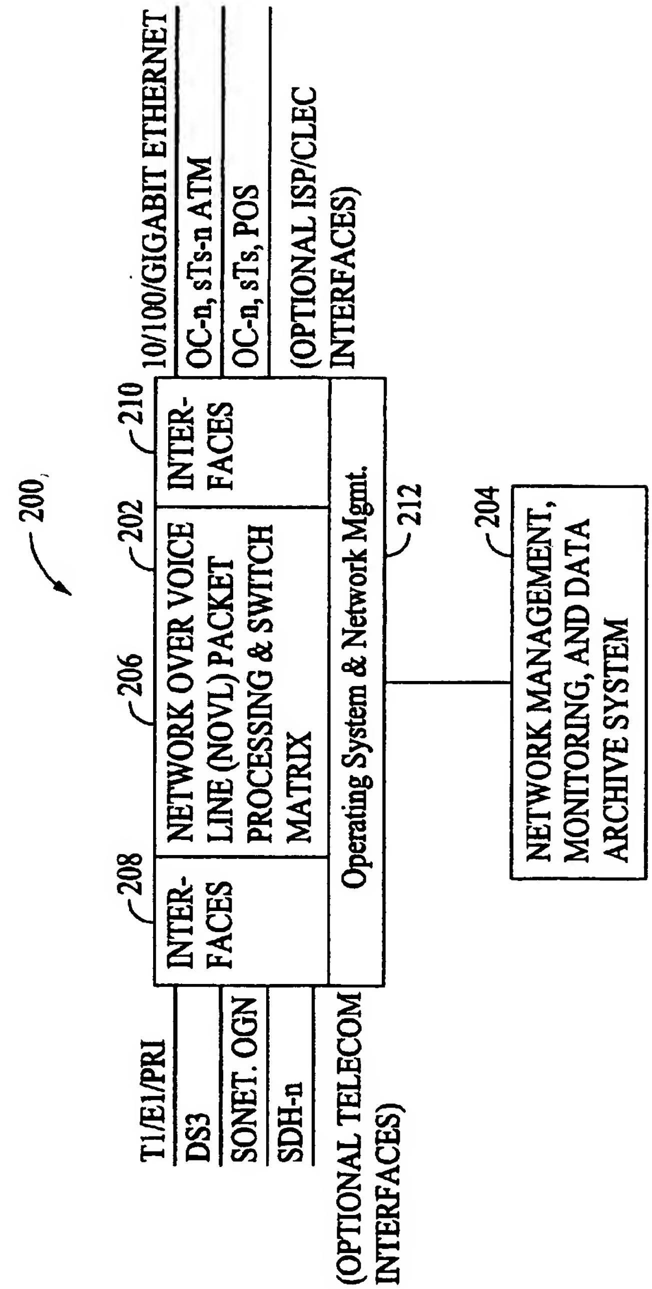
200
202
206
208
210
212
204
T1/E1/PRI
DS3
SONET. OGN
SDH-n
(OPTIONAL TELECOM INTERFACES)
INTER-FACES
NETWORK OVER VOICE LINE (NOVL) PACKET PROCESSING & SWITCH MATRIX
INTER-FACES
Operating System & Network Mgmt.
10/100/GIGABIT ETHERNET
OC-n, sTs-n ATM
OC-n, sTs, POS
(OPTIONAL ISP/CLEC INTERFACES)
NETWORK MANAGEMENT, MONITORING, AND DATA ARCHIVE SYSTEM

FIG. 6

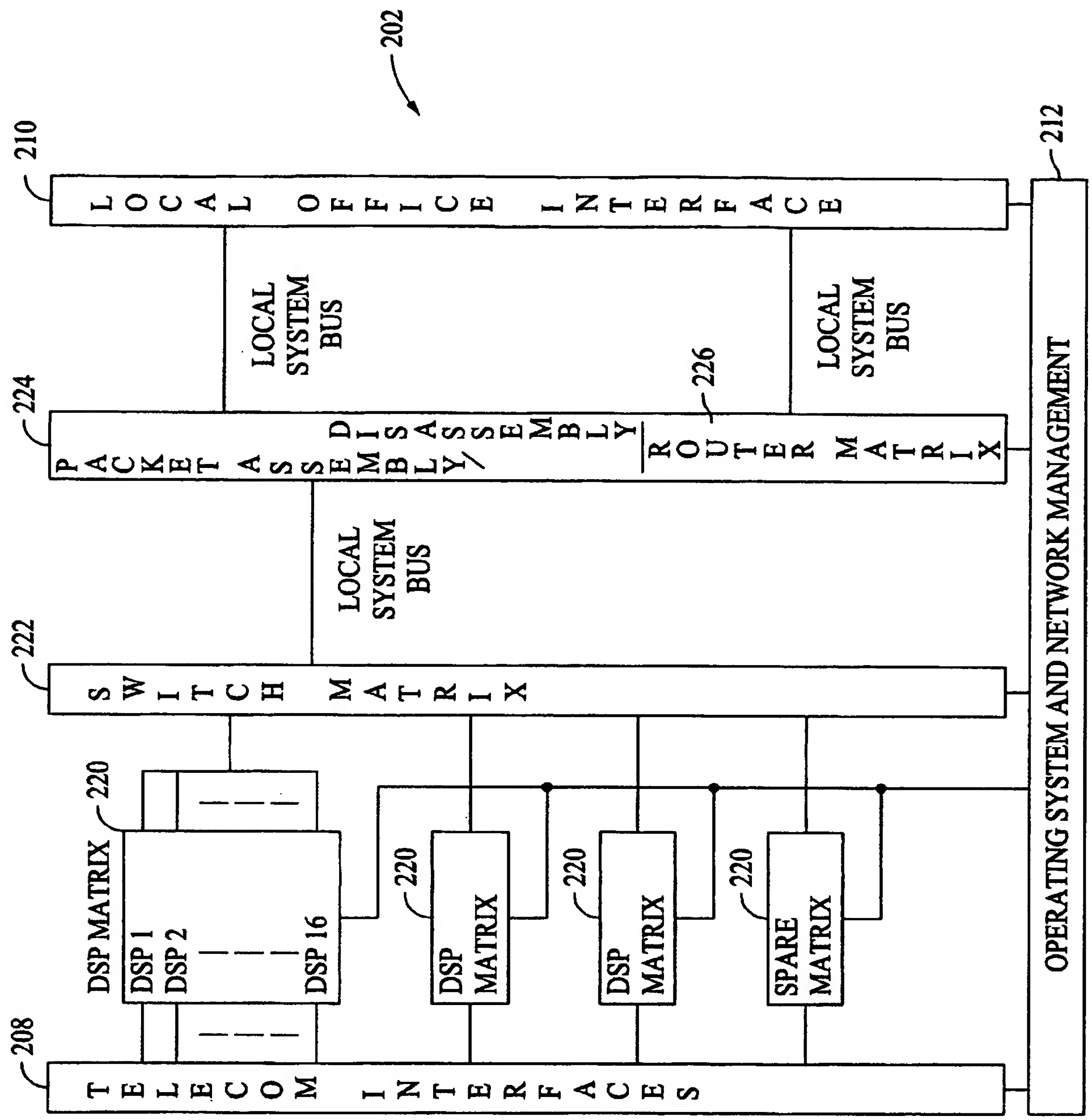
202
210
LOCAL OFFICE INTERFACE
LOCAL SYSTEM BUS
LOCAL SYSTEM BUS
224
PACKET ASSEMBLY/DISASSEMBLY
226
ROUTER MATRIX
LOCAL SYSTEM BUS
222
SWITCH MATRIX
220
DSP MATRIX
DSP 1
DSP 2
DSP 16
220
DSP MATRIX
220
DSP MATRIX
220
SPARE MATRIX
208
TELECOM INTERFACES
212
OPERATING SYSTEM AND NETWORK MANAGEMENT

FIG. 7

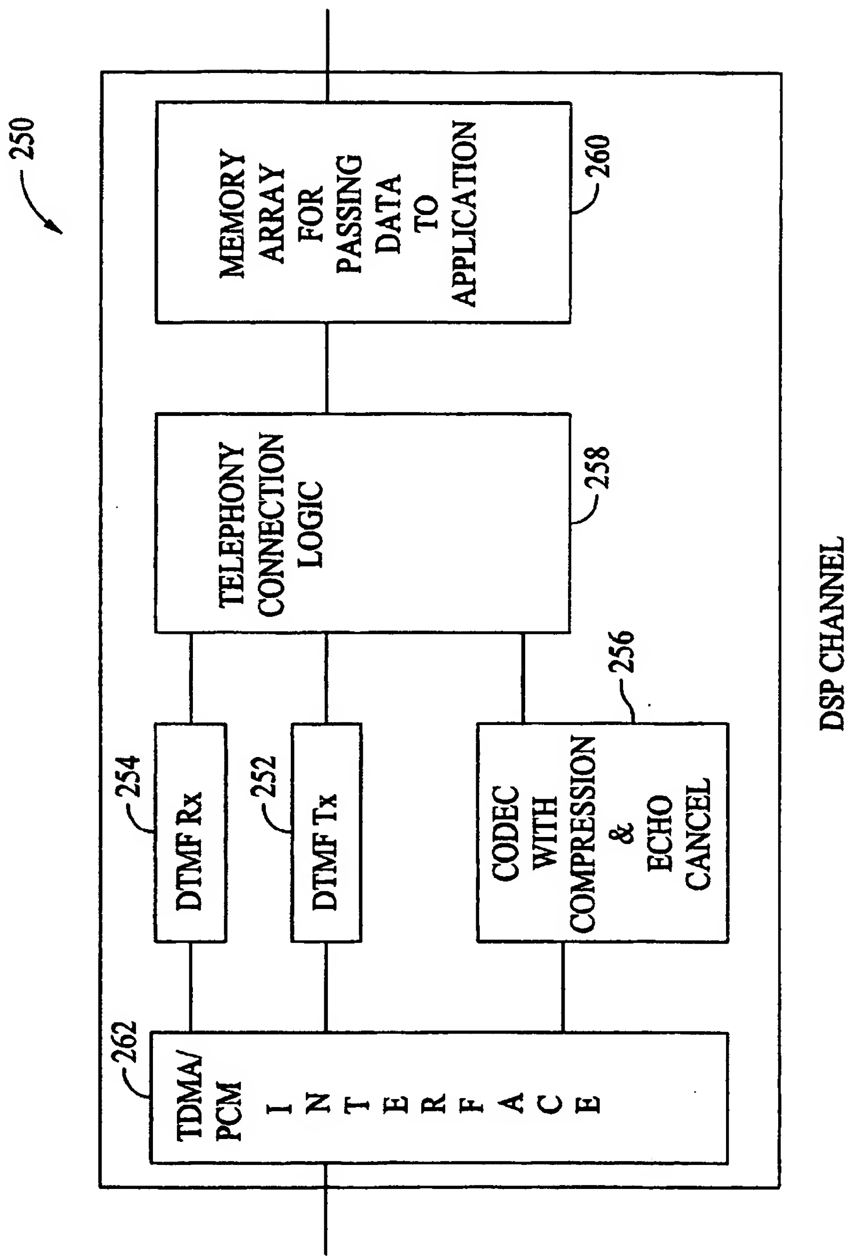
250
MEMORY ARRAY FOR PASSING DATA TO APPLICATION
260
TELEPHONY CONNECTION LOGIC
258
256
DTMF Rx
254
DTMF Tx
252
CODEC WITH COMPRESSION & ECHO CANCEL
TDMA/ PCM INTERFACE
262
DSP CHANNEL

FIG. 8

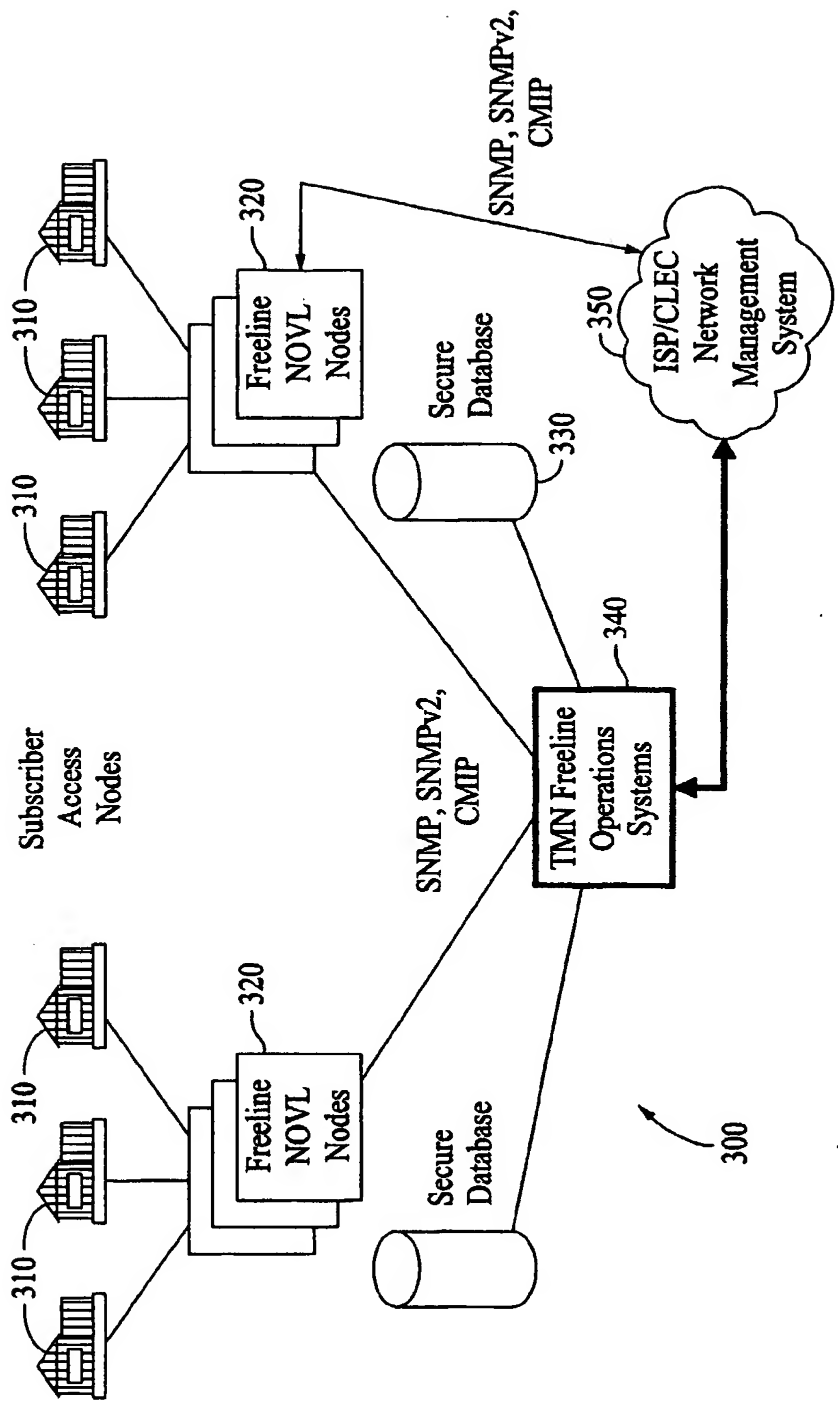
SNMP, SNMPv2,
CMIP
310
310
310
320
Freeline
NOVL
Nodes
Secure
Database
330
350
ISP/CLEC
Network
Management
System
340
TMN Freeline
Operations
Systems
Subscriber
Access
Nodes
SNMP, SNMPv2,
CMIP
310
310
320
Freeline
NOVL
Nodes
Secure
Database
300

FIG. 9

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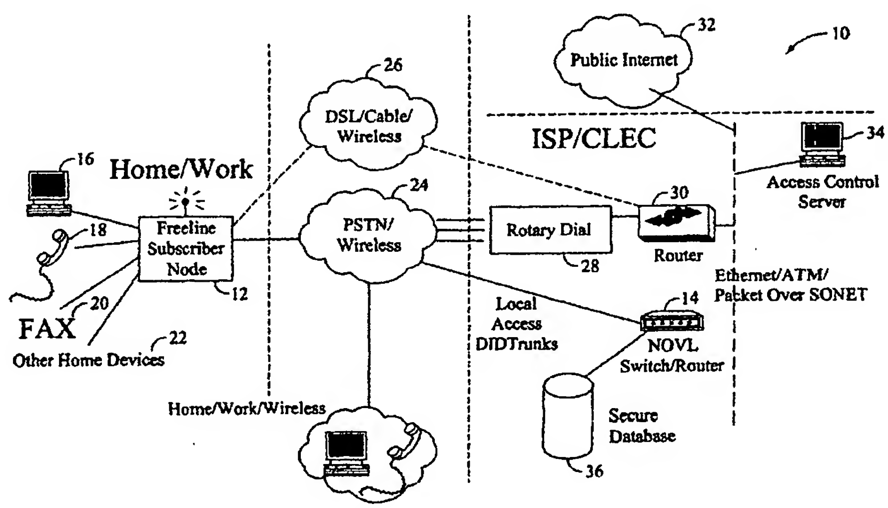
Published:
— *with international search report*

[Continued on next page]

(54) Title: METHODS AND APPARATUS FOR MANAGING AN INTERLEAVED VOICE, VIDEO AND DATA COMMUNICATION SYSTEM



(57) Abstract: In one embodiment, a method for managing a system (10) which provides one or more subscribers (16, 18, 20) with a simultaneous voice and data service is disclosed. The system (10) includes network devices (24, 26, 32) and access devices and the devices are configured to interleave voice, video, and data into subpackets within IP protocol packets for transmission and reception across at least one of public switched telephone networks (PSTN) (24), cable data networks, Digital Subscriber Line (DSL) networks (26), and wireless telephone networks. The method includes remotely provisioning the network devices and access devices within the system (10) through a computer (34) configured for system management and tracking and monitoring system configuration, statistics and service information through the computer (34).

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(88) Date of publication of the international search report: 15 May 2003

For two-letter codes and other abbreviations, refer to the "Guidance Notes on Codes and Abbreviations" appearing at the beginning of each regular issue of the PCT Gazette.

INTERNATIONAL SEARCH REPORT

International application No.
PCT/US02/00990

A. CLASSIFICATION OF SUBJECT MATTER

IPC(7) : H04L 12/28, 12/56
US CL : 370/254, 395.5, 395.52, 401, 402, 420
According to International Patent Classification (IPC) or to both national classification and IPC

B. FIELDS SEARCHED

Minimum documentation searched (classification system followed by classification symbols)
U.S. : 370/254, 395.5, 395.52, 401, 402, 420

Documentation searched other than minimum documentation to the extent that such documents are included in the fields searched

Electronic data base consulted during the international search (name of data base and, where practicable, search terms used)
EAST Database

C. DOCUMENTS CONSIDERED TO BE RELEVANT

Category *	Citation of document, with indication, where appropriate, of the relevant passages	Relevant to claim No.
Y	US 6,044,403 A (GERSZBERG et al) 28 March 2000 (28.03.2000), see the entire document	1-20
Y	US 5,958,016 A (CHANG et al) 28 September 1999 (28.09.1999), see the entire document.	1-20
Y	US 6,125,113 A (FARRIS et al) 26 September 2000 (26.09.2000), see the entire document.	1-20
Y	US 5,970,473 A (GERSZBERG et al) 19 October 1999 (19.10.1999), see the entire document.	1-20
A	US 6,026,086 A (LANCELOT et al) 15 February 2000 (15.02.2000), see the entire document.	1-20

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"&" document member of the same patent family

Date of the actual completion of the international search	Date of mailing of the international search report
29 August 2002 (29.08.2002)	08 OCT 2002
Name and mailing address of the ISA/US Commissioner of Patents and Trademarks Box PCT Washington, D.C. 20231 Facsimile No. (703)305-3230	Authorized officer Toan Nguyen Eugenia Logan Telephone No. 703-305-9600

Form PCT/ISA/210 (second sheet) (July 1998)